USER MANUAL

Maestro

inventis
Audiology equipment

IMPEDANCE-AUDIOMETRY
FUNCTIONALITIES

inventis

MAESTRO

IMPEDANCE AUDIOMETRY FUNCTIONALITIES

USER MANUAL

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Revision: **5**

Date: **10.04.2018**

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CHAPTER 1

Introduction

This manual explains how to conduct audiological tests using the Maestro software application.
The Maestro software is interfaceable with Inventis audiometers and middle ear analyzers connected to a computer by way of a USB port. Depending on the type of instrument

- PC-based
- Hy-tech hybrid
- Stand-alone

a user can control the device when conducting the supported tests and/or transfer test data from the instrument to the Maestro archive.

The graphic interface of the software and the test procedures are described generically in the manual, taking account of the features offered by all Inventis instruments. Headings and/or sections marked with a "(*)" describe functionalities supported only by certain instruments: for a description of the features offered by each instrument, refer to the relative operator manual which also indicates the technical specifications.

CHAPTER 2

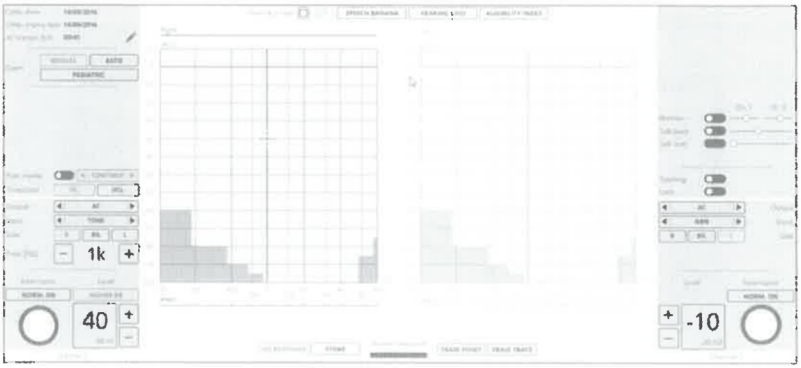
Conducting audiometric tests from a computer

To conduct audiometric tests using a PC-based audiometer or a Hy tech hybrid audiometer in control mode, simply connect the instrument to the computer by plugging in the USB cable and select the test to be carried out.

The following illustration shows the button, located in the bar at the top of the main screen, which is used to activate pure tone audiometry:

Similarly, for every test that can be performed by the connected instrument, the bar displays a button for its selection.

Once an audiological test has been selected (e.g. pure tone), the screen appears as illustrated below:



Whatever the type of test selected, the window will always contain the following five areas:

1. Information area (date of last calibration and calibration expiry date), AC transducer and possible procedures for performing the test currently selected (area at top left of screen). In particular, the section dedicated to the AC transducer allows selection of the type of transducer to be used for air conduction tests: headphone or insert earphone.

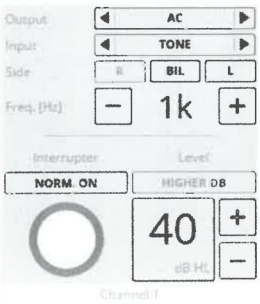
Make absolutely certain that the type of transducer selected here is the same as that effectively being used. If the setting and the transducer are different, the audiometry reading will not be correct.

2. Control panel for channel 1 of the audiometer, referred to typically as the stimulus channel (area at bottom left);
3. Control panel for communication between operator and patient (area at top right)
4. Control panel for channel 2 of the audiometer, referred to typically as the masking channel (area at bottom right);
5. Area displaying the test, containing the graph if any, and controls for saving the test data (screenshot shows the pure tone audiogram and controls for saving data and deleting points).

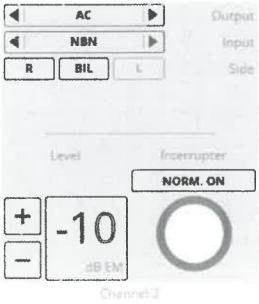
CHANNEL CONTROLS

This section describes the controls for management of the stimulus channel and masking channel. The controls for the two channels appear in mirror image, as shown in the following illustration.

Channel 1 (stimulus)



Channel 2 (masking)



Output. The type of transducer can be selected by clicking the corresponding area with the mouse pointer.
Various transducers can be used according to the type of test, the stimulus or masking channel and the specific instrument:

- air conduction (AC - supra-aural headphones / AC-INS - insert earphones / AC-HF - circum-aural headphones, also suitable for high frequency audiometry)
- bone conduction (BC)
- free field (FF)
- insert type earphones for masking (INS).

The stimulus transducer and the masking transducer can be changed by clicking "O" and clicking "Ctrl + O", respectively.
If a given transducer is found not to be calibrated, the control indicator at the relative output will turn red and it will not be possible to present the tone to the patient.

Input. The type of stimulus can be selected by clicking the corresponding area with the mouse pointer. Various signals can be used according to the type of test, the stimulus or masking channel and the specific instrument. The stimulus signal and the masking signal can be changed by clicking "T" and clicking "Ctrl + I", respectively.

Side. Stimulation and masking can be selected on either side, or both: right (R), left (L) or bilateral (BIL). The stimulus side can also be changed by clicking directly on the audiograms. Remember that by default, selecting one side as the stimulus side in the stimulus channel, the opposite side is assigned automatically to masking; however, the operator can change the masking side, making it the same as the stimulus side. Also, the stimulus side and the masking side can be exchanged with the contralateral one by clicking "TAB" and clicking "Ctrl + TAB", respectively. To set a specific side, on the other hand, click "R", "L" or "B" for the stimulus side and "Ctrl + R", "Ctrl + L" or "Ctrl + B" for the masking side.

Frequency The frequency of the stimulus can be changed in four ways:

- clicking on the corresponding buttons with the mouse pointer,
- using the mouse scroll wheel to adjust the value while keeping the cursor inside the frequency area,
- clicking the desired frequency on the audiogram (possible only if the stimulus and masking functions are not active),
- from the keyboard, by pressing the allocated keys (see heading *General settings*).

Where the concept of frequency is applicable to masking signals, the masking frequency will always be the same as the stimulus frequency. Frequency settings are not possible with speech type audiometry tests.

Intensity. The intensity of the stimulus signal and of the masking signal can be changed in five ways:

- clicking on the corresponding buttons (- and +) with the mouse pointer,
- using the mouse scroll wheel to adjust the value while keeping the cursor inside the stimulus or masking area,
- using the mouse scroll wheel to adjust the value while keeping the cursor on the audiogram currently of interest,
- clicking the desired intensity on the audiogram (possible only if the stimulus and masking functions are not active),
- from the keyboard, by pressing the allocated keys (see heading *General settings*).

For reasons of safety, on reaching an intensity of 100dB HL the screen will show a warning message "HIGHER DB", on which the operator must click in order to raise the decibel level further. The intensity of the stimulus is adjusted by fixed steps of 1, 2, 3 or 5 dB, depending on the selected setting (see heading *General settings*). In addition, it is possible to increase or decrease the intensity of the minimum step available, and this step can also be different to that selected in the settings. This can be done in the following ways:

- holding down the "Alt" key and clicking with the mouse on the + and – buttons,
- using the mouse scroll wheel to adjust the value while keeping the cursor inside the stimulus or masking area and holding down the "Alt" key,
- using the mouse scroll wheel to adjust the value while keeping the cursor on the audiogram currently of interest and holding down the "Alt" key,
- from the keyboard, by pressing the allocated keys (see heading *General settings*).

Interrupter switch. This is used to present the stimulus / masking signal. The signal can be presented by mousing over this button, so as to avoid any extraneous noise, or right-clicking the mouse while keeping the cursor positioned on the audiogram. In addition, the stimulus can be presented from the keyboard, by pressing the allocated key (see heading *General settings*). The masking signal, on the other hand, can be presented using the "AltGr" key.

Norm On. Selecting "Normally On", the operating logic of the interrupter switch will be inverted. The output is always active, and the channel is turned off by operating the switch. The "Normally On" functionality can also be activated/deactivated by clicking the "N" key for the stimulus and the "M" key for the masking signal.

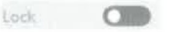
CONTROLS COMMON TO BOTH CHANNELS

Listed below are the controls common to both channels and to the procedure for conducting a test:

Response. Displays the status of the response button: orange Led lights up when the patient presses the button.

Tracking. If TRACKING is activated, the current difference between the intensity of the stimulus and that of the masking signal is maintained constant with any variation in the intensity of the stimulus.

Lock (*). When the lock option is selected, the masking tone will be switched on and switched off synchronously with the stimulus tone. With the lock option selected, the Interrupter and the Normally ON button of the masking channel are deactivated.



OPERATOR - PATIENT COMMUNICATION

Talk over. Allows the activation of communication between patient and operator, either clicking and holding the on-screen button or pressing and holding "T" or "F" on the keyboard (see heading *General settings*). As long as the communication channel remains active, the sending of any other signal

will be inhibited, and then re-enabled only when the TALK OVER button is released. The volume of the communication can be set by sliding the relative cursor, or alternatively turning the scroll wheel of the mouse while keeping the pointer internally of the Talk over area. The levels set in this way are retained every time the module is started up. Depending on the operator-patient communication mode selected (see heading *Audio settings*) the talk over signal can be presented using:

1) a microphone connected to the computer (audio communication in *PC* mode). This functionality is supported only by certain models; for more information, refer to the technical specifications of the instruments.
2) a microphone connected to the "Talk over" input on the rear panel of the instrument (audio communication in *Device* mode)

Talk back (*). This function allows the patient to communicate with the operator. Unlike Talk over, Talk back remains enabled even after the control has been released; to deactivate the function, the operator must press the button a second time. The Talk back function can be activated/deactivated by clicking the "P" key. The volume of the communication can be set by sliding the relative cursor, or alternatively turning the scroll wheel of the mouse while keeping the pointer internally of the Talk back area. The levels set in this way are retained every time the module is started up.

The voice of the patient can be heard:

1) on speakers connected to the computer (audio communication in *PC* mode)
2) through headphones or external speakers connected to the "Monitor" output on the rear panel of the instrument (audio communication in *Device* mode)

Monitor (*). Allows the operator to activate the stimulus and masking signals monitor. The volume of the communication can be adjusted by sliding the relative cursor, or alternatively turning the scroll wheel of the mouse while keeping the pointer internally of the Monitor area. The levels set in this way are retained every time the module is started up. The operator can listen to the monitor signal on speakers connected to the computer (audio communication in *PC* mode) or on headphones/speakers connected to the "Monitor" output of the instrument.

GENERAL SETTINGS

Listed below are the general parameters of the audiometer, which can be accessed from the "*Audiology*" menu.

Settings are divided into the following categories:

- Display options
- Control settings
- Audio settings
- Modes of connection to the instrument

Display options

The operator can select the symbols used to indicate masking for air conduction, bone conduction with and without masking, and UCL threshold. The possible symbols are indicated below, together with the relative default values.

Air conduction

	Right	Left	Bilateral	Default
HL threshold without masking				✓
HL threshold with masking				✓
UCL threshold				✓

Bone conduction

	Right	Left	Default
HL threshold without masking	>	<	✓
	<	>	
	Δ	Δ	
	Λ	Λ	
HL threshold with masking	▷	◁	✓
	◁	▷	
	]	[	
	[	]	

There is also the option of joining bone conduction symbols or otherwise.

Free field

	Right	Left	Bilateral	Default
HL threshold without masking	◀	▶	⋈	✓
HL threshold with masking	S	S	S	✓

Control settings

Indicated below are the control settings common to the various tests.

Default intensity (dB). Default intensity value of the audiometer (between 0 and 60 dB HL). Default value: 40 dB HL.

Maintain intensity. If enabled, the level of the stimulus on the primary channel stays the same (i.e. does not revert to the default value) when changing the stimulation side, output, input or frequency. Default value: No.

Maintaining the previous intensity of the stimulus when changing frequency, transducer or stimulation side can result in potentially harmful signals being presented to the patient.

Step dB (*). Allows selection of the incremental step for the attenuators, which can be 1, 2, 3 or 5 dB, depending on the instrument in use. Default value: 5.

Extended FF intensity range (*). The intensity range for free field audiometry tests, in other words the maximum output values; can be normal or extended. Default value: No (not extended).

Beep on patient response. Enables or disables the beep signal generated when the patient response button is pressed. Default value: No.

Second response button (*). Selects whether or not the audiometer is equipped with two response buttons (right / left). Default value: No.

Shortcut keys. The table below lists the shortcut keys that can be used when conducting a test.

Increasing intensity direction	
Decrease stimulus intensity (by minimum step)	Down arrow (ALT + Down arrow)
Increase stimulus intensity (by minimum step)	Up arrow (ALT + Up arrow)
Decrease masking intensity (by minimum step)	CTRL + Down arrow (CTRL + ALT + Down arrow)
Increase masking intensity (by minimum step)	CTRL + Up arrow (CTRL + ALT + Up arrow)
Decrease frequency	Left arrow
Increase frequency	Right arrow
Send stimulus	ENTER
Stimulus continuous on/off	N
Masking continuous on/off	M
Send masking	AltGr
Change stimulus side as	TAB
Change masking side	CTRL + TAB
Set left side on stimulus	L
Set left side on masking	CTRL + L
Set right side on stimulus	R
Set right side on masking	CTRL + R
Set bilateral side on stimulus	B
Set bilateral side on masking	CTRL + B
Change stimulus input	I
Change masking input	CTRL + I
Change stimulus output	O
Change masking output	CTRL + O
Talk over	T
Talk back	P
Erase point	DEL
Store point	SHIFT
Speech player play	F5
Speech player stop	ESC
Word scoring	Left arrow / Right arrow

The operator can select the direction of changes in intensity of the stimulus (increase/reduce), and customize the keys for presenting the stimulus, and for storing the current point and the word count.

Audio settings

The mode of managing audio communication can be selected both for the reproduction of speech material used in speech audiometry tests, and for operator-patient communication.

Operator - patient communication

- DEVICE mode: operator and patient communicate by way of a microphone connected to the Talk over input of the instrument. The voice of the patient can be heard through headphones or external speakers connected to the "Monitor" output of the instrument.

- PC mode (*): audio communication occurs by way of the USB cable. The operator microphone for talk over and live speech must be connected to the audio input of the computer, whereas in order to hear the talk back signal and the monitor signal, either headphones or speakers must be connected to the audio output of the PC.

Default value: DEVICE mode.

Reproduction of speech material

- Ext mode: in this configuration, in order to present the speech material stored on the computer to the patient, the audio output of the PC must be connected to the EXT input of the audiometer. To conduct a "live" speech test, a microphone must be connected to the MIC input.

- USB mode (*): speech material stored on the PC is reproduced by way of the USB cable

Default value: EXT mode.

Depending on whether or not the instrument supports USB mode, the relative microphones and head/earphones can be selected so as to ensure correct operation of the audio connection.

Modes of connection to the instrument (*)

In the case of audiometers supporting the HY-TECH hybrid functionality, the operator can choose whether to control the instrument from the PC (using

mouse, keyboard or touch screen) or to connect the instrument to the PC for the purpose of transferring test data.

ABOUT THE INSTRUMENT

This screen presents general information on the connected instrument:

- Model
- Serial number
- Licenses activated (*)
- Information regarding technical services.

In particular, the owner or operator can activate licenses available for the instrument in question (such as optional tests). Any activation of licenses will be memorized by the instrument. The window shows all the licenses available for the model of instrument connected to the computer. To activate a particular test, click or tap the corresponding ENABLE button and enter the license code. Users can obtain the code by contacting their dealer or technician, providing the serial number of the instrument on which the license is to be activated.

In the case of instruments that can be controlled from a computer, it is also possible to activate a test by selecting it directly from the list displayed on screen.

CHAPTER 3

Pure tone audiometry

Pure tone audiometry is the standard hearing test: it allows the ENT specialist to determine the type and extent of hearing loss at different frequencies and consequently provides the basis for diagnosis and care of the patient.

HEARING THRESHOLD

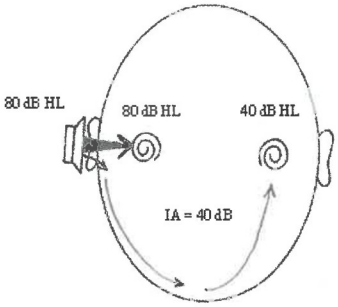
The *hearing threshold* is a basic concept in the field of audiology. Related to a stimulus, the hearing threshold is the minimum intensity that will elicit an auditory response.
In clinical hearing assessment, more exactly, this is referred to as the *audiometric threshold*, that is, the weakest level of a stimulus signal that the patient is able to hear at least 50% of the time.
In practice, pure tone audiometry is the test that determines the audiometric threshold of the patient at different frequencies.
The tone signals generated by the audiometer, and the level of the audiometric threshold, are measured in decibels (dB). This particular level is referred to as dB HL (Hearing Level): in contrast to the dB SPL (Sound Pressure Level), which represents the actual intensity of the sound, the dB HL includes a compensating factor whereby a person with normal hearing tested under ideal conditions will have a hearing threshold of 0 dB HL at every frequency. These compensating factors, which are different for each frequency and each type of transducer, have been calculated from population based studies and are published in international standards (EN ISO 389-1, 389-2, 389-3, 389-4, 389-5 and 389-7).

MASKING

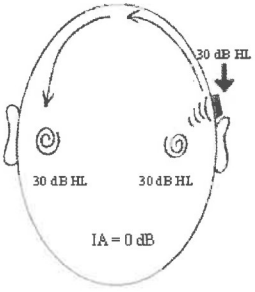
When a signal is presented to one ear, the cochlea of the other ear can also be stimulated, in dissimilar measure, as a result of the sound being transmitted through the bones of the skull. The mechanism whereby a signal presented to one ear stimulates the cochlea of the opposite ear is known as *cross hearing*. In situations where cross hearing could occur, a masking signal must be used.

Masking consists in raising the threshold of the contralateral ear temporarily by presenting an appropriate masking signal (generally a narrow band noise, NBN, centred on the current stimulation frequency) at a predetermined level. This prevents the contralateral ear from sensing the stimulus presented to the ear being tested, which could lead to the responses of the patient registering false positives and affecting the measured threshold.
The unit of measure for narrow band masking noise (NBN) is dB EM (*Effective Masking*). In contrast to dB HL, the correction factor introduced into dB EM is such that when using a given stimulation frequency (e.g. 1 kHz), a narrow band noise of predetermined intensity (e.g. 20 dB EM) will render inaudible any signal of intensity less than or equal to the noise in question (or in the case of the example, any tone of 1 kHz frequency generated at an intensity of up to 20 dB HL).

A fundamental concept applied in the selection of a suitable masking strategy is that of interaural attenuation (IA): the reduction of energy in a sound signal presented to one ear and transmitted by bone conduction to the contralateral ear.



Interaural attenuation with AC



Interaural attenuation with BC

For air-conducted signals, the level of IA varies according to the frequency and the transducer adopted (supra-aural headphones or insert earphones) and, to an extent, from one individual to another.
In the case of headphones, reference can be made to the following table, taken from Goldstein BA, Newmann CW (1994) "Clinical masking: a decision making process", in Katz J ed. "Handbook of Clinical Audiology".

Frequency (Hz)	*125*	*250*	*500*	*1000*	*2000*	*4000*	*8000*

IA (dB)	35	40	40	40	45	50	50

In the case of insert type earphones (ER-3 and 5 models), IA is much higher (above 70 dB).
In the case of bone conduction signals, on the other hand, IA is near to 0 dB: consequently, one can never be certain which cochlea has been stimulated by a BC tone, irrespective of where the vibrator is positioned, which means that cross hearing can always be a factor in bone conduction.

Having introduced the concept of interaural attenuation, instances of cross-hearing can be identified with the following formula, in which case a masking signal becomes necessary:

$$\mathbf{AC_{TE} - IA \geq BC_{NTE}}$$

where: $\mathbf{AC_{TE}}$ = air conduction threshold for the Tested Ear
$\mathbf{IA}$ = interaural attenuation
$\mathbf{BC_{NTE}}$ = bone conduction threshold for the NonTested (contralateral) Ear

In the same way as for a test tone, if the masking noise is of sufficiently powerful intensity, it can be transmitted by bone conduction to the tested ear. If the intensity of the masking noise is sufficient to mask not only the non-tested ear but the tested ear as well, the effect is referred to as *overmasking*. The maximum intensity of masking noise above which overmasking (M_{max})occurs is given by this formula:

$$\mathbf{M_{max} = IA + BC_{TE}}$$

Accordingly, overmasking occurs when the level of the masking signal (EML_{NTE}) is as determined by this formula:

$$\mathbf{EML_{NTE} \geq IA + BC_{TE} + 5}$$

AIR CONDUCTION AUDIOMETRY

The general rule when performing an audiometric test with air-conducted signals is to use a pure tone input, which can be either continuous or pulsed. Referring to the considerations of the previous section, the operator must decide whether or not a masking signal is needed, and determine its level.
To ensure a dependable outcome, the test should be conducted in suitably quiet surroundings, and preferably in a soundproof booth. If there is no such enclosure available, the use of special sound-isolating headphones is recommended, in particular Audiocups, which can be supplied as an optional

accessory with audiometers. Standard EN ISO 8253-1 specifies the maximum permissible ambient noise level compatible with dependable results.
With the patient wearing the headphones, the test will be best conducted employing the procedure described below, which is taken from EN ISO 8253-1.

INSTRUCTIONS TO THE PATIENT

Tell the patient to press the button (or raise a hand) when a sound is heard, and release the button (or lower the hand) only when the sound is no longer heard.

PRESENTATION AND INTERRUPTION OF TEST TONES

The test tone should be presented for a duration of 1 or 2 seconds. When there is a response from the patient, the interval between successive test tones must be varied, although never reduced to a duration less than that of the actual tone.

INITIAL FAMILIARIZATION

Before determining the hearing threshold, the patient must be familiarized with the task by presenting a signal of intensity sufficient to elicit a precise response. After this familiarization step, the operator can be confident that the patient understands the procedure and is able to respond correctly.
The following method of familiarization can be used:

1. present a 1000 Hz tone at a clearly perceptible level, say, 40 dB HL for a person with normal hearing;
2. reduce the level of the tone by steps of 20 dB until there is no longer any response;
3. increase the level of the tone by steps of 10 dB until a response is elicited;
4. present the tone again at the same level.

If the responses are consistent with the presentation of the tones, the familiarization process is complete. If not, the procedure should be repeated. Following a second failure, repeat the instructions.
In cases of profound deafness, these procedures may not be applicable.

ESTIMATION OF HEARING THRESHOLD (ASCENDING METHOD)

Conduct the test initially on the ear that the patient considers to be the better ear. The different frequencies should be tested in the following sequence: 1000, 2000, 4000, 8000, 500, 250 Hz, including half octaves if appropriate. Perform the following sequence of steps.

1. Present the first tone of the test at a level 10 dB less than the lowest level of the response given by the patient during the familiarization session. Following each tone that fails to elicit a response, increase the level of the test tone by steps of 5 dB until a response is triggered.

2. Following the response, reduce the level by degrees of 10 dB until there is no response, then begin the ascending sequence again. Continue until the patient produces three responses at the same level on a maximum of five ascending sequences.
   Using the ascending method, if less than three responses at the same level are elicited on five ascending sequences (or less than two responses on three ascending sequences, using the short method), present a tone at a level 10 dB higher than the level of the last response. Repeat the test procedure.
   It has been shown that a shortened version of the ascending method gives comparable results, and in certain instances can be considered suitable for the purpose in question. Using the short method, continue with the test until the patient gives at least two responses at the same level on a set of three ascending sequences.

3. Move on to the next frequency, starting at a level that ought to be audible on the basis of the previous responses, and repeat step 2. Complete the test on one ear, at all the frequencies, in the sequence indicated.
   Repeat the measurement at 1000 Hz. If the results of the repeat test at 1000 Hz on the same ear are comparable with the first, registering a difference of less than 5 dB from the previous level, then proceed with the other ear. If there is an improvement of the order of 10 dB in threshold hearing, repeat the test at other frequencies in the same order until the values reflect a deviation equal to or less than 5 dB.

4. Continue until the test has been completed on both ears.

BONE CONDUCTION AUDIOMETRY

Given that bone-conduction IA is near to 0 dB, the exact determination of the hearing threshold for an ear using bone-conduction audiometry requires masking of the non-tested contralateral ear at all levels.
Follow the steps indicated below to establish the threshold level.

1. Position the bone vibrator on the patient's mastoid, behind and as near as possible to the conch, without touching, and place the masking headphone over the non-tested ear (the unused headphone should be positioned forward of the tested ear, leaving the cavity exposed). Make certain that the head bands of the two transducers do not interfere with one another.
2. Begin the test with the ear the patient considers to be the better ear, at a frequency of 1000 Hz.
3. Establish the threshold level without the masking signal, using the shortened ascending method described in the previous section. Bear in mind that the result of this operation cannot serve as a true estimate of the bone-conduction threshold, given the possibility that the non-tested could be affected by occlusion (see next section).
4. Activate the masking noise and establish the correct intensity of the masking component — hence the hearing threshold — using the Plateau method described in the next section.
5. Move on to the other frequencies, following the same sequence as indicated for air-conduction testing. Continue until the test has been completed on both ears.

HOW MASKING IS APPLIED

As mentioned previously, whenever cross-hearing could be an issue, a masking noise must be applied to the non-tested ear, delivered by way of the headphones or a dedicated insert earphone.
The method described here for determining the level of the masking tone is the so-called 'plateau' technique described by J.D. Hood in 1960. This involves gradually increasing the masking noise applied to the non-tested ear in such a way as to arrive at a plateau — that is to say, a range of masking values across which there is no increase in the hearing threshold of the tested ear. The presence of the plateau suggests that, within the range of masking intensity values in question, there will be no undermasking (i.e. a level not sufficient to prevent cross-hearing) and no overmasking.

The plateau technique involves the following three steps.

1. Find the threshold for the ear being tested, without masking.
2. Activate the masking noise at an initial level of 10 dB higher than the air conduction threshold for the non-tested ear (in the case of bone conduction, see the following note); except in certain particular cases, this level will not produce overmasking.
3. Check the threshold of the tested ear again. If the threshold of the tested ear has not changed, increase the intensity of the noise by steps of 5 (or 10) dB, checking the value after each increment; if there is no change in the threshold over three successive increases in the intensity of the noise (or two increases, for steps of 10 dB), this means that the plateau has been found, and the value can be considered as the threshold. Conversely, if there is a change in the threshold value, this indicates that it was influenced by undermasking, and the procedure must be repeated with the new threshold level.

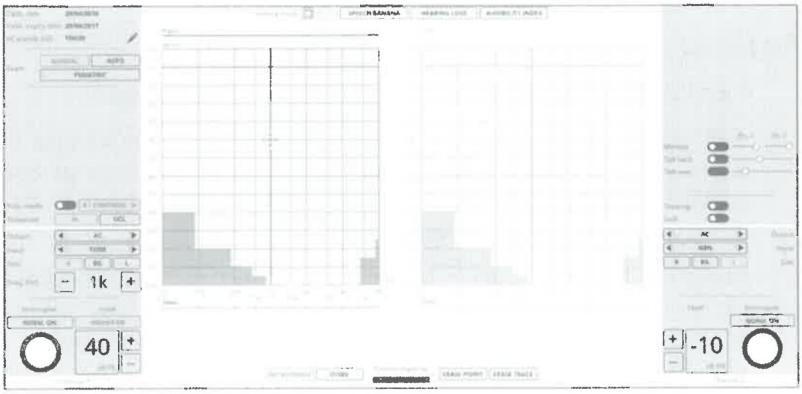
If bone-conduction is being used for the test, the initial level of the masking signal should be increased to compensate for the *occlusion effect* (OE). OE — due to the headphone being placed over the non-tested ear — causes an increase of the bone-conduction tone threshold in the frequency range between 250 and 1000 Hz; in particular, the threshold in question increases by 15 dB at 250 Hz and 500 Hz, and by 10 dB at 1000 Hz. The initial value of the masking noise therefore needs to be higher than the AC threshold of the non-tested ear, by 25 dB at 250 Hz and 500 Hz, and by 20 dB at 1000 Hz.

UCL AUDIOMETRY

The UCL (Uncomfortable Level) test is used to determine the sound intensity at which acoustic stimuli become intolerable to the patient. This information is important in establishing the upper limit of the patient's hearing range, typically for the purpose of fitting hearing aids, as well as for detecting instances of recruitment.

PURE TONE AUDIOMETRY WINDOW

Once the pure tone audiometry test has been selected, the screen appears as illustrated below:



In addition to the common controls described in the previous chapters, the pure tone audiometry screen shows:

Type of test:

- Manual
- Auto: selects automatic audiometry (see heading *Automatic audiometry*)
- Pediatric (*): selects video-VRA audiometry for children (see heading *Video-VRA pediatric audiometry*)

Pulsed mode:

- Continuous
- Single pulse of programmable duration (see heading *Settings for pure tone audiometry*)
- Pulsed at a rate of 0.5 Hz, 1 Hz, 2 Hz, or operator-settable rate

Consult the technical specifications of the instruments to find out the pulsed modes supported.

Threshold:

- HL
- UCL

Store. Memorizes the threshold at each frequency. The instrument can be set in such a way that the test moves on to the next frequency automatically, whenever the threshold is saved. The same operation can also be performed via the keyboard, pressing the allocated key (see heading *General settings*). For details on configuration of the automatic frequency jump, see heading *Settings for pure tone audiometry*.

Not heard. Where the stimulus is presented at maximum intensity and the patient does not respond, the threshold level can be labelled with a different symbol. In particular, a down arrow is added to the bottom part of the symbol.

Canc. Point. Removes the symbol on the audiogram corresponding to the selected side, transducer and frequency. The operator can also remove the current data item by pressing "Canc" on the keyboard.

Canc. Trace. Removes the entire current trace.

Display mode. The operator can decide whether to display the right and left side audiograms superimposed or separately, clicking on the relative button positioned above the audiogram. It is also possible to add a mask to the background of the audiogram, which can be:

- Audibility index
- Speech spectrum
- a grey scale corresponding to the level of hearing loss, with variations at 15 dB, 25 dB, 40 dB, 55 dB, 70 dB and 90 dB (scale taken from JG Clark (1981), *Use and abuse of hearing loss classification.* ASHA;23: 493-500)
- Letters and phonemes (referred to English language).

Audiogram. Certain areas of the audiogram will appear highlighted; these indicate frequency / intensity values not obtainable with the selected transducer. Left-clicking on the pure tone audiogram, frequency and intensity are set at the value corresponding to the point selected with the mouse pointer.

A pure tone audiometry can be superimposed on the pure tone audiometry in progress, or on pure tone tests saved previously. This option can be activated by selecting the menu item *Audiology → Superimpose audiometry* and identifying the pure tone audiometry to superimpose, or by dragging the desired audiometry from the *session data management bar* in the main area of the graphic interface (see *Maestro user manual*). The superimposed pure tone audiometry will also be shown on the printout of the test.

Once the test has been saved, it can be displayed on a screen other than the control screen. Not only does the control screen show the display modes indicated in the control window, it can also be made to show:

- The audiogram classification (Merluzzi or MPB)[1] or

[1] Enrico Pira, Davide Bosio, Franca Merluzzi: *Prevention of hearing loss due to noise in the workplace – Guidelines proposed by the Italian Society of occupational medicine and industrial hygiene.*

- The average of the pure tones (PTA): here, the operator can set the frequencies on which the average is calculated (default frequencies 500 Hz, 1000 Hz and 2000 Hz).

Settings for pure tone audiometry

Listed below are the specific parameters for pure tone audiometry, accessible from the menu "*Audiology → Control settings*", Pure tone audiometry section.

Automatic frequency jump. Enables or disables automatic transition to the next frequency after a threshold value on the audiogram has been saved (using the Store function). Default value: No.

Frequency jump mode: return to 1000 Hz. Allows selection of the next frequency value, once the maximum value of 8k Hz has been reached. If return to 1000 Hz is activated, the frequency reverts to 1 kHz, if not, it will default to 125 Hz. Default value: Yes.

Frequencies to examine. Allows selection of the set of frequencies to be examined. Bear in mind that the 1 kHz frequency cannot be deselected. Default value: all frequencies.

Single pulse duration. Duration of the pulse in "single pulse" mode. Default value: 1000 ms.

Custom pulsed period. Duration of period in pulsed tone presentation mode with customized pulse duration. Default value: 1500 ms.

Keep masked and non masked BC traces separate. The operator can save both masked and non masked Bone Conduction traces in the audiogram. Default value: No.

AUTOMATIC AUDIOMETRY

Auto Threshold pure tone audiometry allows the operator to establish the hearing threshold for different frequencies by way of an automatic procedure. The method applied is the Hughson-Westlake technique, as adapted by Martin, which provides a modified and shortened version of the method indicated in ISO standard 8253-1 (described in previous sections) for determining the AC threshold without masking.

For selection of the frequencies that will be examined when conducting the test, refer to the heading *Settings for pure tone audiometry*. Bear in mind that the 1 kHz frequency cannot be deselected.

To begin the test, simply press the *START* function button. The test finishes automatically once all the enabled frequencies have been analyzed for both ears. The test can in any event be terminated by pressing the *END* function button.
After pressing *START*, a message will appear warning the operator that any air-conduction related threshold data saved previously will be erased.

The automatic test comprises an initial familiarization step serving to train the patient in the threshold identification procedure (the caption *FAMILIARIZATION* is displayed on the screen during this step). The stimulus presented to the patient is generated at a frequency of 1 kHz and an initial intensity of 40 dB HL (which cannot be changed). The patient must press the response button every time the stimulus is heard, and release the button thereafter. In particular, the stimulus is presented for a duration of 1.7 seconds, followed by a pause of random duration between 1.7 seconds and 2.5 seconds.
The adopted method of determining the threshold envisages an increase of 5 dB in the level of the stimulus, and a reduction of 10 dB. The threshold is established as the level for which at least 2 positive responses are elicited from the patient over 3 sequences.
Once the threshold has been identified, the familiarization step concludes and the test will start automatically.

The first frequency examined in the Auto Threshold sequence is 1 kHz. Once the hearing threshold has been identified for this frequency, the test passes on to the next frequency. The scanning order for examination of the frequencies selected previously is as follows: first, frequencies higher than 1 kHz, in ascending order, then frequencies lower than 1 kHz, in descending order, and finally, the 1 kHz frequency again.
Once all these frequencies have been analyzed for one ear, the test is repeated automatically on the other ear.
In practice, if the patient is unable to hear a stimulus signal even at the maximum permissible level for a given frequency and after it has been presented 10 times, the signal in question will be identified as "not heard".
The audiogram will be updated in real time during the course of the test, to show the results obtained.

VIDEO-VRA PEDIATRIC AUDIOMETRY

Pediatric audiometry involves the use of visual stimuli (reinforcements), consisting in practice of up to three LCD displays (one in front and one on

either side of the patient), on which images or video clips are shown during the test. The test can be conducted adopting the two conventional modes, namely VRA (*Visual Reinforcement Audiometry*) and CPA (*Conditioned Play Audiometry*), described below.

CPA (Conditioned Play Audiometry)

This procedure is used with children over 2-3 years of age, and does not differ greatly from that of familiar audiometric tests. A stimulus is presented, and the child presses the button when the sound is heard.
When a correct response is given (i.e. the button is pressed after the stimulus has effectively been presented) the child receives a "reward", which can be depicted by an image or a film clip shown on the display. The purpose of the rewards is to ensure that the child will stay keenly attentive and continue to cooperate during the test.

VRA (Visual Reinforcement Audiometry)

Very small children, especially those with a significant auditory deficit, tend to be extremely uncooperative; consequently, an ear test using CPA is unsuitable in this situation. In these cases, Visual Reinforcement Audiometry is used.
The VRA procedure includes an initial training *step in which the child is presented with a clearly audible sound stimulus, followed by a visual stimulus (referred to as a* reinforcement). This creates a conditioned reflex in the child, who on hearing the sound will turn in the direction of the reinforcement, expecting the reward.
At this point the operator can start the actual test, which involves presenting sounds and rewards in alternation, and using the reactions of the child in search of the reward as an indication that the sound has been heard; in this way, the hearing threshold is determined for each frequency.

Conducting the test

To conduct a pediatric audiometry test, select the relevant type of pure tone audiometry, Pediatric.
The controls used in conducting the test are the same as for standard pure tone audiometry, except for the selection of CPA or VRA mode and the dedicated visual reinforcements panel.

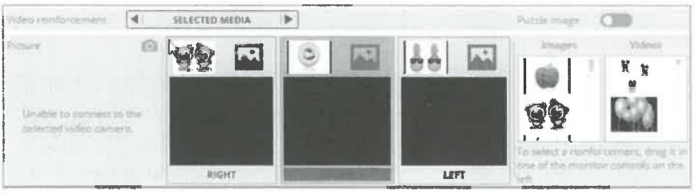
The operator can switch from one mode to the other simply by selecting the CPA button or the VRA button. The only difference between the two tests is the manner of presenting the visual reinforcement: in the case of the VRA test, it is the operator who decides when to activate the reinforcement,

whereas with CPA, the reinforcement is activated automatically when the patient presses the response button following presentation of a stimulus.

For VRA tests, to facilitate the operations of activating and deactivating the reinforcement, the reinforcement can be controlled by way of the patient button used in the conventional audiometric test. The button used by the child during a CPA test uses the same channel as the auxiliary patient response button.

MANAGING VIDEO REINFORCEMENTS

The central part of the screen, illustrated below, is dedicated to the management panel for video reinforcement media.



Visual reinforcements can be sent to one, two or three displays and consist in images or video clips. The software is supplied with a series of images and clips already installed; to add others, refer to the following section on *Settings for pediatric audiometry*.
There are three ways of presenting a video reinforcement: the operator can use the same image or clip every time ("Selected media") or random pictures and animations ("Random image" and "Random clip").

In the case of a CPA test, if the video reinforcement associated with the monitor is an image, this can be presented either in its entirety, or one piece at a time in the manner of a jigsaw puzzle ("Puzzle image").
The central part of the window shows what is currently visible on the patient monitor, plus a small preview (always on) of the image / clip selected currently for each of the three displays.
The number of the displays represented in the lower part of the screen must be the same as the number of monitors actually placed in front of the patient. One display is highlighted on the screen — corresponding to the monitor on which the visual reinforcement is presented — and will depend on the selection made in the pediatric audiometry settings window (see heading *Settings for pediatric audiometry*).
In particular:

- in the case of the VRA test, the active monitor can be made to coincide with the stimulus side (if the monitor relative to the current side is not enabled, the existing monitor will remain selected), or selected by the operator, clicking on the display;
- in the case of the CPA test, the active monitor is always the one selected in the settings window.

The right hand part of the panel shows the lists of images and clips that can be used as video reinforcement. To use these rewards, simply use the *drag and drop* functionality to transfer the image or clip to one of the monitors shown in the central part of the panel. If the "Random image" or "Random clip" option is selected, the list of rewards will be disabled, and images or clips cannot be selected individually.

CAMERA

The left part of the panel shows the live feed from the webcam trained on the child, if the device is connected and configured (see heading *Settings for pediatric audiometry*). The camera serves a dual purpose: the first is that of allowing the operator to observe patient reactions even with eyes fixed on the screen and unable to look at the child directly; the second is that of allowing the operator to take snapshots of the child (pressing the relative button), which can then be used as reinforcements and projected onto LCD monitors. In effect, the pictures taken are added to the list of reward images.

Settings for pediatric audiometry

Listed below are the specific parameters of pediatric audiometry, accessible from the menu "*Audiology → Pediatric audiometry settings*".

Reinforcement time Indicates the duration for which the reinforcement is activated, variable between 1 and 10 seconds (can always be cut short manually, whatever the setting).

VRA mode. The following parameters apply to VRA mode only. Selection of monitor to be used: the operator can choose between two options, namely "ipsilateral", in which case the selection of the game will be determined automatically and correspond to the side on which the stimulus is presented currently, or "selectable from the display". With the VRA test, the operator can also set the mode in which the reinforcement activation button will be used: reinforcement active only when the button is pressed and held; reinforcement activated by pressing the button, and terminated automatically at the end of the set time, or manually by pressing the button again.

CPA*mode.* The following parameters apply to CPA mode only. The operator can select the monitor to be used, and it is also possible to set the time limit for a response on the part of the patient, at between 1 and 5 seconds from the end of the stimulus. Accordingly, the response is considered valid (and the reinforcement activated automatically) if the patient presses the button while the tone is active, or within the set number of seconds after the tone has finished.

Reinforcements. The two lists displayed at the bottom of the window contain images and clips that can be used as video reinforcement. An image or a clip can be removed from list by selecting the name and clicking the icon; similarly, new items can be introduced using the button.
The films used are *.mp4, .mpg, .mpeg* files, widely available on the web. Images can have *bmp, jpg* or *png* extensions.

Number of monitors. Indicates the number of patient monitors utilized. The minimum setting is 1, the maximum setting 3. Default value: 3.

Positioning of monitors. Identifies the position of each monitor relative to the patient (left of patient, right of patient, directly in front).

Use video source. Enables / disables the use of a video camera to picture the child in the VRA module. If enabled, the video source selected in the Maestro general settings will be used. Default value: No.

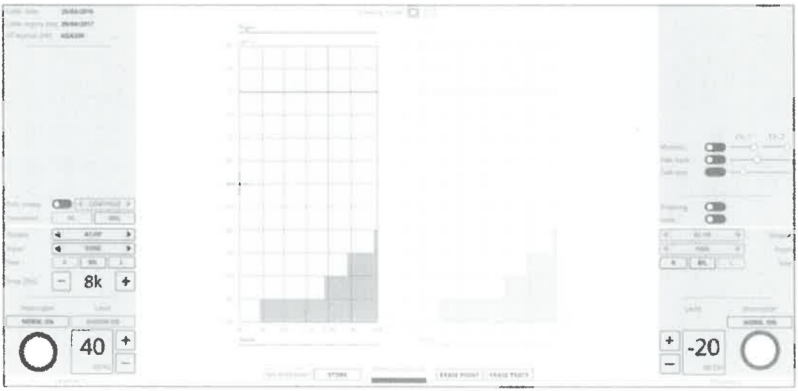
Limit movement of mouse to primary monitor. Enables / disables the possibility of limiting the movement of the mouse to the primary monitor, i.e. that of the operator. Default value: No.

CHAPTER 4

High frequency pure tone audiometry

High frequency pure tone audiometry is used to find the hearing threshold at frequency values from 8 kHz up to 20 kHz (16 kHz in free field).
Analysis conducted in this frequency range allows the identification of hearing damage caused by exposure to noise, or by treatments requiring the use of ototoxic drugs. In effect, such impairments tend to occur earlier at these frequencies than at traditional frequencies.

The following illustration shows the high frequency audiometry screen.



The controls used for the test are the same as for tests conducted at standard frequency, except for the audiograms, which in this instance cover the higher band of frequencies.

SETTINGS FOR HIGH FREQUENCY PURE TONE AUDIOMETRY

Maximum frequency HF: 20000 Hz. Sets the upper frequency limit for the HF pure tone test. Possible values are 16 kHz or 20 kHz. Default: No (hence 16 kHz).

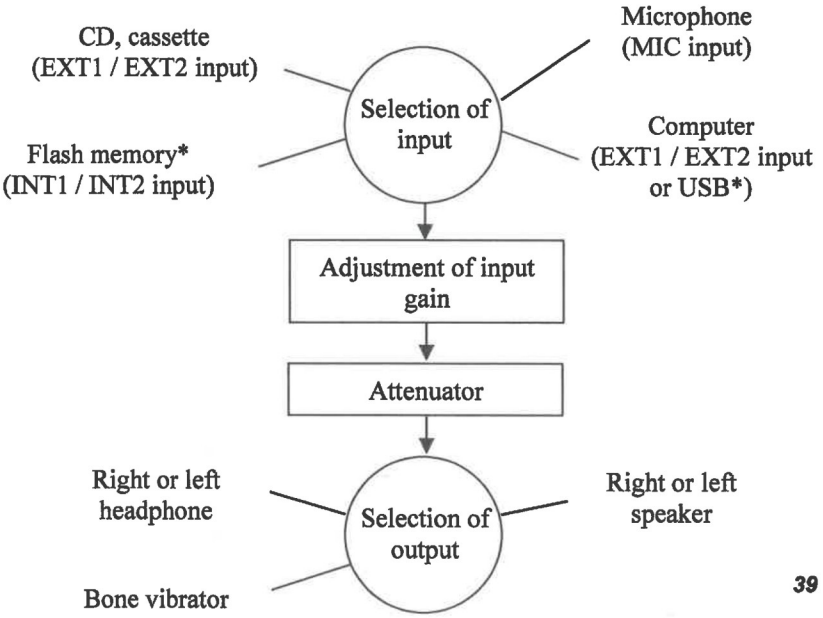
Frequencies of 18 and 20 kHz are available in air conduction modes only if enabled in the settings. Because stimulus signals at these frequencies are not standardized, their intensity cannot be expressed in dB HL, only in dB SPL. Consequently, these frequencies are not displayed on the audiogram.

CHAPTER 5

Speech audiometry

Pure tone audiometry provides only partial information on the abilities of the patient to hear and understand the spoken voice, the most important of audible signals. Accordingly, more information can be gained by conducting a speech audiometry test, in which the patient is presented with a succession of standardized speech stimuli, typically single syllables, words or phrases. The purpose of the test is to measure certain aspects of the patient's hearing capability, in particular verbal intelligibility, i.e. the ability to recognize input signals as sound stimuli that have meaning. The speech material can be presented directly by the operator over a microphone, recorded on a CD or a cassette tape and transmitted from a suitable player connected to the EXT input of the audiometer, or stored on the computer and transmitted to the instrument by way of a USB* cable or via cables connected to the EXT inputs.

The following diagram illustrates how the speech material is presented to the patient via the audiometer.



Selection of input

The input signals for speech audiometry vary according to the particular instrument, and include:

- Microphone (MIC input): the test can be conducted in "live speech" mode using a microphone connected to the MIC input on the rear panel of the instrument (if the communication mode selected is classic mode – see heading *Audio settings) or a microphone connected to the audio input of the computer (if the communication mode selected is* USB*);
- External audio source (EXT1/EXT2 inputs): it is possible to reproduce the signal from external audio sources (CD or cassette player) connected to the EXT1/EXT2 inputs on the rear panel of the instrument;
- Speech material stored on PC: speech material stored on a computer can be reproduced by selecting inputs EXT1/EXT2 and transmitting the audio signal via cables connected to the EXT1/EXT2 inputs on the rear panel (if the communication mode selected is classic mode) or selecting inputs USB1/USB2 and transmitting the audio signal via the USB cable (if the communication mode selected is *USB**);
- Speech material stored in the flash memory* of the instrument: in the case of instruments that support an internal flash memory, speech material can be reproduced by selecting inputs INT1/INT2.

Adjustment of input gain

Before starting a speech audiometry test, the level of gain on the selected input must be adjusted. The purpose of this operation is to ensure that recorded speech material utilized for the test, or the voice of the operator in the case of a live speech test, will be heard at the correct volume.

Where recorded speech material is used, the input gain will be correct if, applying a calibration tone supplied with the material, the loudness measured on the VU-meters of the two channels — displayed on the screen of the computer — reaches 0 dB, that is to say, the point at which the meter changes from green to red.

In the case of a live speech test using the MIC input, the VU-meter of channel 1 should register 0 dB on average when the words are pronounced, keeping at a fixed distance from the microphone.

To set the gain of the stimulus and masking signals, simply adjust the position of the relative cursor in the dedicated sections of Channel 1 and Channel 2.

The gain value of each input is memorized and set again at the same level every time the same type of input is used thereafter.

Attenuator

The speech signal is adjusted to the level set by the operator (the level of channel 1 or of channel 2, depending on whether the signal is a stimulus signal or a masking signal).

Selection of output

The speech signal can be presented to the patient through headphones (AC), on loudspeakers (FF), or by way of the bone vibrator (BC), depending on what outputs are supported by the particular instrument. In the first two instances, the operator can select right, left or bilateral as the stimulation side, whereas in the case of the BC output, it is not possible to select bilateral as the stimulation side.

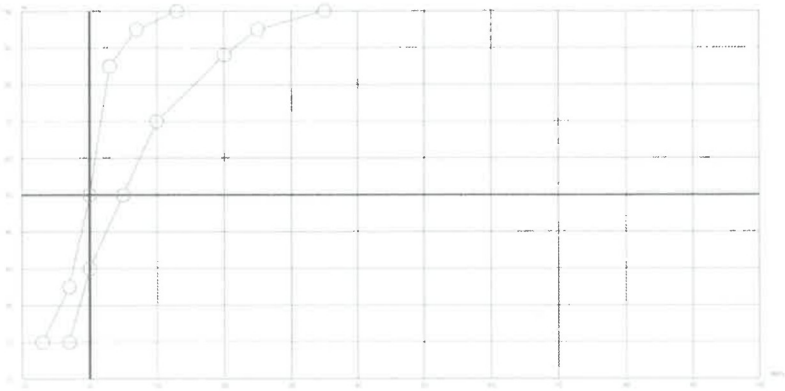
MASKING AND COMPETING SIGNAL

As in pure tone audiometry, a masking signal can be used in speech audiometry to avoid the eventuality of measurement errors being induced as a result of cross-hearing. The masking signal is generated through channel 2 of the audiometer; the recommended signal for speech audiometry is *Speech Noise* (SN), which produces an energy output concentrated in the frequency band typical of the human voice. It is also possible to select white noise (WN) as the masking signal.

To assess the ability of a patient to recognize speech in an environment with background noise, in other words under conditions typical of everyday life, speech audiometry can be conducted by presenting a *competing sound* either to the tested ear or to the contralateral, non-tested ear, or to both. The competing sound, which is always generated through channel 2 of the audiometer, can be speech noise or white noise, or a signal incorporated into the audio track of the selected speech material, recorded typically on the channel opposite to that containing the word lists.

ARTICULATION FUNCTION AND SRT

The articulation function (or performance-intensity function) is the curve that describes the speech recognition score (average percentage of correct answers) as a function of speech intensity level. The articulation function provides information on how speech material of different types (monosyllabic words, spondaic words, etc.) is identified correctly by the patient in relation to the intensity of presentation.
The following illustration provides an example of articulation functions.

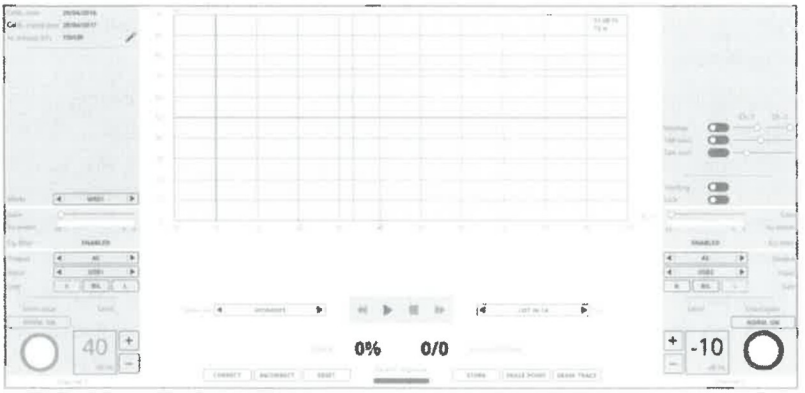


The two curves in the graph relate to two different types of speech material, and more exactly, lists of spondees (bisyllabic words with meaning) and of logotomes (bisyllabic nonsense words, phonetically balanced). The level of intensity at which 50% of correct responses are obtained with lists of bisyllabic words is known as the *Speech Recognition Threshold* (SRT). The average SRT for adults with normal hearing is 20 dB SPL; standard EN 60645-2 defines this level as equivalent to 0 dB HL. In the case of speech audiometry, accordingly, to convert a value of intensity expressed in dB SPL to the corresponding value in dB HL, simply subtract 20 dB from the value.

The articulation function in persons with normal hearing presents a steep curve: on average, 100% of correct responses will be registered at a value as low as 27.5 dB SPL (or 7.5 dB HL)

SPEECH AUDIOMETRY WINDOW

Once the Speech test has been selected, the screen appears as illustrated below:



Apart from the common controls described in the previous chapter, the speech audiometry screen presents a number of additional functions described under the following sub-headings.

Test modes

Allows selection of the type of speech audiometry, from Word Recognition Score (WRS), Speech Reception Threshold (SRT), Most Comfortable Level (MCL), and Uncomfortable Level (UCL). In WRS mode, the system can acquire and store data relative to two different speech signals when selecting modes WRS1 and WRS2, respectively.

Also, enabling the relative option from the settings menu, the operator can activate the "WRS single point" function: in this case the WRS1 and WRS2 modes allow the system to save one percentage only of correct responses for a single value of intensity of the stimulus signal. In this configuration WRS1 and WRS2 data will be presented in a table without the aid of the audiogram.

Binaural mode

If "WRS single point" mode has been enabled and the current outputs are air conduction or free field, the *Binaural* button can be used to select the same input for both channels. In this mode, accordingly, the stimulus can be presented to the right ear and the left ear simultaneously and at different intensities.

Percentage of correct answers

Using the *Correct*, *Incorrect* and *Reset* buttons, the operator can determine the percentage of correct answers given by the patient. More exactly, the *Reset* button zero-sets the percentage value defined currently, whilst the *Correct* and *Incorrect* buttons, depending on the word count mode selected

from the settings menu (see heading "Settings for speech audiometry"), can be used by the operator as follows:

"Fixed number of words" mode. The number of words to be presented to the patient is fixed (see heading "Settings for speech audiometry") and the percentage of correct answers is calculated simply by pressing the *Correct* button when the patient gives the right answer. The *Incorrect* button decreases the number of correct answers by one, and is used only in the event of an answer being identified wrongly as correct.

"Variable number of words" mode. The number of words to be presented is not fixed, but determined during the course of the test. To calculate the percentage of correct answers, press the *Correct* button for a right answer and the *Incorrect* button for a wrong answer: in practice, the *Incorrect* button increases only the total number of words presented to the patient, whereas the *Correct* button increases both the total number of words and the number of right answers.

The percentage of correct answers can be calculated not only using the *Correct/Incorrect* buttons, but also by way of the computer keyboard, pressing the allocated keys (see heading *General settings*).

When the option *Phonemes mode* is enabled (see heading "Settings for speech audiometry") the system will calculate the percentage of phonemes correctly identified. To save the number of phonemes correctly identified for each word, the operator must use the number buttons "0", "1", "2" and "3" on the panel illustrated below

Store/Delete

Using the *Store* button, the percentage of correct answers given by the patient for each dB value of the stimulus, or the threshold level in SRT/MCL/UCL mode, can be memorized on the graph. The score and/or threshold can also be stored using the relative allocated key on the computer keyboard (see heading *General settings*).
To delete a previously stored point from the graph, position the pointer on the specific dB value and press the allocated key on the keyboard (see heading "General Settings"), or use the *Delete* button. The *Delete Trace* button can be used to eliminate all memorized points associated with the current trace.

To eliminate threshold values stored in SRT/MCL/UCL modes, select the mode in question and remove using the *Delete* or *Delete Trace* button.

VU-meters and Gain

The VU-meters positioned near the channel controls indicate the loudness of the source applied as the input signal. It is fundamentally important to set the level of the VU-meters correctly, so that the test will give a credible result (see heading "Adjustment of input gain"). The level of the input signal can be adjusted by positioning the cursor of the relative gain control, using the scroll wheel of the mouse while keeping the pointer within the Gain area.

Speech material

The speech audiometry test can be conducted using the voice of the operator (live speech) or playing prerecorded speech material.
If the pre-recorded material is stored on the computer, it can be indexed. It is possible to define the list of words contained in each audio track: during reproduction of the audio tracks, the current word is displayed in the control panel relative to the speech material, in such a way that the operator can monitor the words presented to the patient without having to listen to the audio signal.

In the event that speech material stored on the computer is being used, the operator must deactivate all audio effects installed on the computer (e.g. Dolby) and set the sound card volume to maximum.

The indexing of audio tracks can be accomplished with the aid of Inventis ATIT software, included in the contents of the CD supplied with the instrument. The folder of the computer where the speech material is stored contains as many subfolders are required for each of the different types of speech material (e.g. *monosyllabic words* or *spondaic words*). Each sub-folder contains the tracks and the lists of words to be reproduced (*.wav* files).

Located in the central area of the speech audiometry window are the controls for reproduction of the speech material, which appear as illustrated below:

The following table describes the operation of the buttons used for reproduction of the lists of speech material:

	Material Allows selection of the speech material to be utilized, from among all the possible folders in the directory (enabled only when playback is not currently active)

	List - Allows selection of the specific list to be reproduced, from the selected folder of speech material (enabled only when reproduction is not currently active)
	Play or key "F5" Activates reproduction of the speech material.
	Pause or key "F5" Allows the operator to suspend playback of the speech material.
	Stop or key "ESC" Terminates reproduction of a list.
	Rewind If speech material is currently playing and the content is indexed, repeats the last word pronounced.
	Forward If speech material is currently playing and the content is indexed, moves on to the next word.

If the speech material is indexed, the word (or phrase) being pronounced is displayed above the reproduction controls.

Settings for speech audiometry

Phonemes mode. Enables calculation of the percentage of phonemes correctly identified in each of the words presented.

Word count mode. Allows the operator to select either of the two modes of storing the correct/incorrect answers given by the patient.

- Variable number of words: the operator keeps a tally of both correct and incorrect answers; accordingly, the current score reflects the total number of words presented.

- Fixed number of words: the operator keeps a tally of correct answers only; accordingly, the current score reflects the total number of words in the list.

Number of words in list. Where "Word count mode" is set to "fixed number of words", allows selection of the number (1 to 99) of words making up each list.

Enable WRS single point. If enabled, allows calculation of a single numerical value when using speech audiometry in WRS mode. If the WRS single point option is enabled and the outputs currently selected are air conduction or free field, then Binaural mode can be activated.

*Pause after each word. If the speech material is indexed, enabling this option allows the operator to suspend playback of the list automatically (*Pause) after each word.

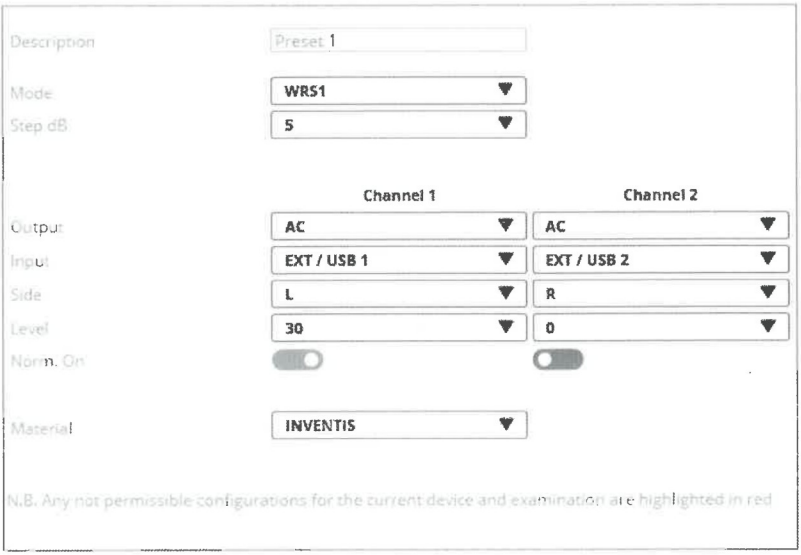
Enable EQ filter for AC. Enables / disables the equalization of the speech audiometry output signal when conducting an air conduction test with TDH-39 headphones.

Show standard curves. The operator can view a maximum of two normal hearing reference curves on the audiometry graph. The "Standard curve 1" and "Standard curve 2" sections allow the operator to perform actions on each of the two curves: enable viewing, set the name of the associated speech materials, and define the points (maximum of 5 points on each curve).

Speech audiometry presets

The following section describes the functionality of speech audiometry preset, which is accessible from the menu "*Audiology → Speech audiometry preset*".

In effect, the user can create, change and even delete the start-up presets for speech audiometry. These presets consist in a group of settings, including the status of the channels and the speech materials, to be applied when conducting speech audiometry, as illustrated below:



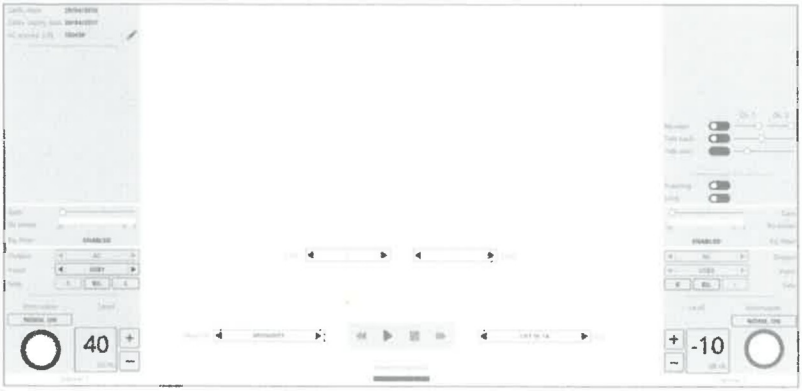
If a preset is selected, the information will appear in the speech audiometry window.

CHAPTER 6

Master Hearing Aid

The Master Hearing Aid (MHA) test is used to assist the selection of a hearing aid for patients with impaired hearing. With the MHA test, the audiometer can simulate the use of a hearing aid by filtering the speech signal sent to the patient; in this way it is possible to verify and quantify any improvement in speech recognition attributable to the introduction of a hearing aid.
Various different filters can be selected and activated independently for each channel. These are high-pass type filters with amplification of 6 dB, 12 dB, 18 dB and 24 dB per octave.

The screen for the MHA test is illustrated below:



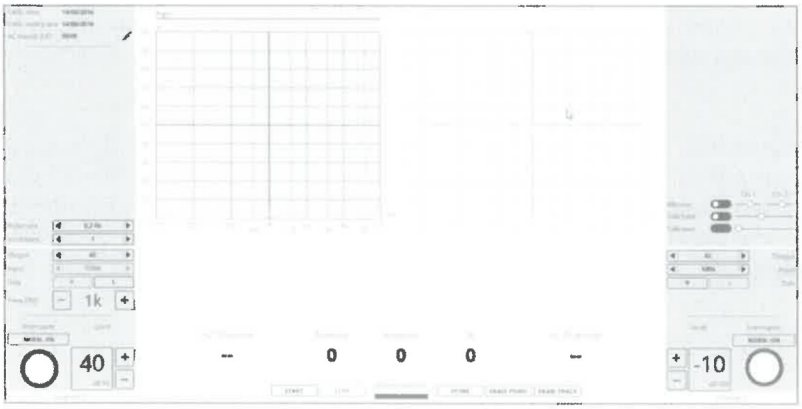
In addition to the controls already described under "Speech audiometry", the central part of the MHA test window shows the filters active on each channel. Clicking on the corresponding area, the operator can select the type of filter to be activated on channel 1 and 2, respectively, or disable the filter.
The MHA test does not include any save or store function, consequently there is no printout available.

CHAPTER 7

SISI (Jerger)

The Short Increment Sensitivity Index (SISI) test assesses the ability of the ear to identify brief variations of intensity in a continuous pure tone.
As a rule, the test is conducted by presenting a continuous pure tone at different frequencies, selecting an intensity 20 dB above the hearing threshold identified previously for the ear and at the test frequency, introducing small increments in intensity of short duration and asking the patient to press the button each time an increment is perceived.

The SISI test screen is illustrated below.



The panel on the left contains the common controls, as well as controls allowing the operator to change the amplitude of the increments, and the frequency with which the increments are presented.

The amplitude of the increments can be set at 0.2, 0.4, 0.6, 0.8, 1, 1.5, 2, 3, 4 and 5 dB; the increments of 3, 4 and 5 dB should be used only for the initial familiarization stage of the test, during which the responses given by the patient to the stimuli received will not be counted. For more information on

the amplitude of increments permissible for different instruments, refer to the technical specifications of the particular audiometer.

Possible frequency values for presentation of the increments are 0.2 Hz, 0.5 Hz and Random (between 0.2 Hz and 0.5 Hz, changing at each increment).

The area below the graph contains essential information relative to the progress of the test, namely the number of increments presented (Stimulus), the number of responses made by the patient to the stimuli (Response), and the percentage of responses to the total number of stimuli generated (%). Also indicated are the threshold levels found during pure tone audiometry at the current frequency and for the current output (audiometry associated with the SISI test).
The central part of the screen is taken up by the SISI graph, which shows the percentage of increments detected by the patient at each frequency analyzed.

The SISI test is an automatic procedure. Once the frequency and the intensity of the tone have been set, the test is set in motion by pressing the START function button. A sequence of 20 increments is presented to the patient, during which the responses are counted only if the button is pressed within one second of the increment occurring. The analysis of the selected frequency ends automatically once the 20 increments have been presented; the sequence can however be aborted by pressing the END function button.

Increments of 1 dB should be imperceptible, or no more than barely perceptible, to patients with normal hearing or with transmission or retro cochlear hearing loss; accordingly, if the percentage of increments perceived by the patient is higher than 30%, this could be an indication of cochlear disorder.

CHAPTER 8

QuickSIN

The difficulty of understanding speech against background noise is a very common source of discomfort in patients affected by hearing loss. The measurement of SNR loss (reduction of signal to noise ratio) is fundamentally important, given that the ability to comprehend speech in noisy conditions cannot easily be assessed on the sole basis of a pure tone audiometry test (Killion & Niquette, 2000).
The QuickSIN™ test allows a rapid assessment of SNR loss using phrases that are presented to the patient together with background noise consisting of four voices all speaking at the same time (*4-talker babble noise*). The lists of QuickSIN™ speech material consist in six phrases, each containing 5 key words, which are presented to the patient with a signal-to-noise ratio of 25, 20, 15, 10, 5 and 0 respectively. SNR loss is assessed by allocating a score for each one of the six phrases presented, calculated by attributing one point for each key word correctly identified.

The QuickSIN™ speech material can be stored in the internal memory of the instrument (if available), which is selectable by setting inputs INT1/INT2, or in the computer, selectable by setting inputs EXT1/EXT2 or USB1/USB2 depending on the audio mode selected (see heading "Audio settings"). If the folder containing the QuickSIN™ speech material is not available, the test cannot be conducted.

The QuickSIN™ material includes:
- Calibration tone at 1-kHz
- 12 standard lists (*Standard*): phrases and 4-talker babble noise recorded on the same channel with preset SNR levels from 25 dB to 0 dB.
- Practice lists A, B and C (*Practice*): used for the purpose of familiarizing the patient with the test. These lists are not the same as the standard lists and cannot be used for calculating SNR loss.
- 12 "split-tracks" standard lists: phrases and babble noise recorded on two separate channels. These lists can be used for verifying the effectiveness of directional hearing aids. The test can be conducted employing two different procedures:

- Subjective: the level of the stimulus signal and that of the background noise can both be set manually. By getting the patient to change position, the operator can give a demonstration of the benefits provided by directional hearing aids. It is not possible to calculate SNR loss in subjective mode.
- Objective: once the level of the stimulus signal has been set, the level of the background noise is set automatically, so that the SNR will be reduced progressively from 25 dB to 0 dB. In objective mode, it becomes possible to calculate SNR loss and compare the values obtained by getting the patient to change position

- 12 HFE lists (high-frequency emphasis): lists recorded using filters that boost high frequencies by 30 dB.
- 12 HFE-LP lists (high-frequency emphasis low pass): lists recorded in such a way as to emphasize high frequencies, and applying a low pass filter with cut-off frequency of 3 kHz. These lists, played in combination with the HFE lists, can be employed to assess the benefits deriving from the use of hearing aids with high frequency emphasis.

The screen for the QuickSIN™ test is illustrated below:

When conducting the QuickSIN™ test, in addition to the common controls for the two channels, the operator can use the control panel for the reproduction of speech material (described under the heading "Speech audiometry window"). The specific functions of the QuickSIN™ test are described below.

Reproduction of QuickSIN™ material

Using the relative buttons of the section providing control over reproduction of the speech material, the operator can select the specific QuickSIN™ material and list to utilize. Having selected the list in the QuickSIN™ test window, the six phrases it contains will be displayed on the screen, with the key words highlighted in uppercase.

Pressing the *Play/Pause* button starts the playback of the list: the phrase currently being presented is displayed on the screen. During playback of the list, the operator can move from one phrase to the next, or to the preceding phrase, using the *Forward/Rewind* buttons, Each phrase is presented to the patient with a signal to noise ratio that decreases progressively by steps of 5 dB, starting from 25 dB and reducing ultimately to 0 dB.

Saving data

At the end of each phrase, playback of the list is paused automatically, and the operator can set the relative score using the "0", "1", "2", "3", "4" and "5" buttons. Pressing *Play*, the next phrase of the list will be presented to the patient.

At the end of the test, if a score has been given for each of the six phrases, the operator can calculate the SNR loss and save the value by pressing *Store*. The following table indicates the criteria for interpreting the value of SNR loss obtained:

SNR LOSS	*DEGREE OF LOSS*	*EXPECTED IMPROVEMENT WITH DIRECTIONAL HEARING AID*
0-3 dB	Normal / almost normal	Able to hear better than a person with normal hearing in noise
3-7 dB	Slight SNR loss	Able to hear almost as well as a person with normal hearing in noise
7-15 dB	Moderate SNR loss	Directional microphones can help. Consider microphone array
>15 dB	Severe SNR loss	Significant improvement of SNR required. Consider FM systems

Selection of outputs and gain adjustment

The lists of QuickSIN™ material can be presented to the patient through headphones (AC) or in free field (FF). For patients with an average pure tone threshold (PTA) $\leq$ 45 dB HL, it is advisable to set the level of the stimulus signal at 70 dB HL. For values of PTA $\geq$ 50 dB HL, set the signal at a level the patient considers to be "sufficiently high".

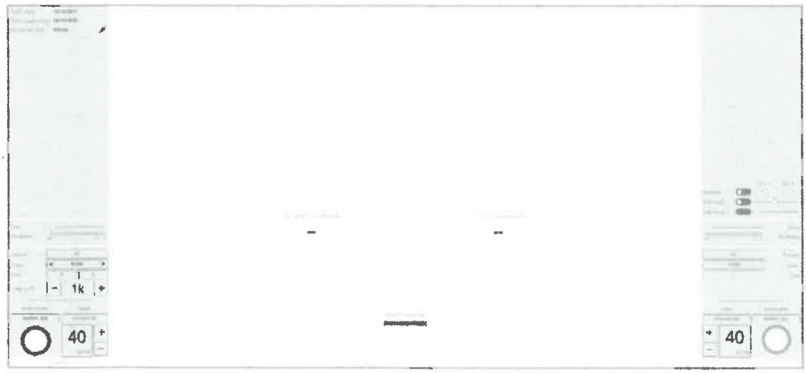
Before starting the QuickSIN™ test, the level of the signals must be adjusted using the 1-kHz calibration tone, which is included in the folder of QuickSIN™ material. To set the level by way of a VU-meter, refer to the heading "Speech audiometry window".

CHAPTER 9

Stenger

The Stenger test can be used to establish whether or not a measured hearing threshold is credible. This is based on the "Stenger effect": if tones of the same frequency but different intensities are presented to the two ears of a patient simultaneously, only the stimulus in the ear that receives the tone of higher intensity will be heard.
Having established the hearing threshold for a given frequency, the Stenger test consists in presenting a tone at 5-10 dB higher than the threshold to the ear with apparently normal hearing, and presenting the same tone at 5-10 dB lower that the presumed threshold to the ear which the patient claims is affected by hearing loss. If the hearing loss is real, the patient will indicate that the signal can be heard in the good ear. A different response means that the hearing loss is not genuine.
The Stenger test can be conducted using pure tone signals or speech material.

The screen for the Stenger test is illustrated below:



In the left hand panel, the user can select the stimulus side associated with channel 1, which corresponds to the ear in which the patient claims to have a loss of hearing. The side opposite to that selected for channel 1 is set automatically to channel 2.

Selecting a pure tone or a warble signal as the input, the central area of the test window is used not only to monitor the response of the patient (already described in the heading "Conducting audiometric tests by computer"), but also to display the threshold levels established previously for the right ear and the left ear during the course of the pure tone audiometry test, for the frequency and output currently selected — audiometry associated with the Stenger test. The stimulation frequency can be set using the control located in the left hand panel.

Conversely, if the intention is to conduct the test using speech material, then operator simply selects the input from EXT1/EXT2, INT1/INT2 and USB1/USB2 (if the material presented to the patient is prerecorded) or MIC/MIC PC (if the test is to be conducted in "live speech" mode), depending on what inputs are supported by the instrument with which the test is being carried out, and on the audio mode selected (see heading "Audio settings"). In this instance, the central area of the test window no longer shows the threshold levels detected using pure tone audiometry; instead, the controls for the reproduction of speech material will appear. Moreover, when conducting the test with speech material, the level of the source used as the input can be set by way of the VU-meters positioned near the controls for the two channels (see heading "Speech audiometry window").

The stimulus is always switched on and off synchronously for the two channels, so that the signal can be presented to or withheld from the two ears simultaneously.

The Stenger test does not include any save or store function, consequently there is no printout available.

CHAPTER 10

Live functionality and transfer of tests

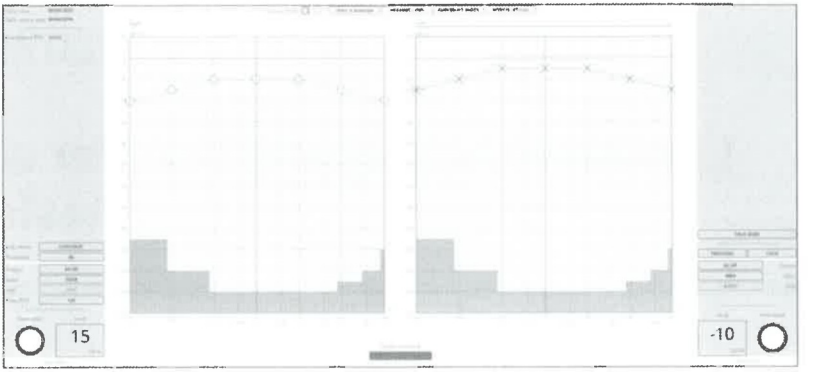
To access the function allowing transfer of test data, and the *live* functionality if required, click the button identifying the connected instrument, which will be found on the bar at the top of the main screen. For example:

Make certain that the mode of connection to the instrument is as indicated in the menu Instrument → PC (see heading *General settings*).

LIVE FUNCTIONALITY

The *live* functionality allows the operator to follow a test currently in progress on the computer screen, as well as on that of the instrument. This facility is available only for pure tone audiometry, pediatric audiometry, speech audiometry, tympanometry and acoustic reflex tests.

The following illustration shows the *live* window that will appear in the case of a pure-tone audiometry test.



The window shows:

- the status of channel 1 (level, side, input, output and frequency);
- the status of channel 2 (level, side, input, output);
- the status of the response button (orange Led lights up when the patient presses the button);
- the presentation mode of the stimulus signal (continuous or pulsed);
- the current threshold (HL or UCL);
- the audiogram in progress.

Display mode. The operator can decide whether to display the right and left side audiograms superimposed or separately, clicking on the relative button positioned above the audiogram. It is also possible to add a mask to the background of the audiogram, which can be:

- Audibility index
- Speech spectrum
- a grey scale corresponding to the level of hearing loss, with variations at 15 dB, 25 dB, 40 dB, 55 dB, 70 dB and 90 dB (scale taken from JG Clark (1981), *Use and abuse of hearing loss classification*. ASHA;23: 493-500)
- Letters and phonemes (referred to English language).

Audiogram. Certain areas of the audiogram will appear highlighted; these indicate frequency / intensity values not obtainable with the selected transducer.

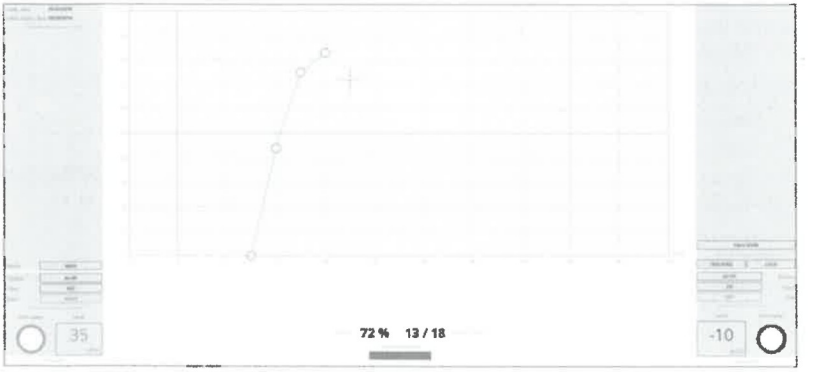
A previous exam can be superimposed on the current audiogram. This option can be activated by selecting the menu item *Audiology → Superimpose audiometry* and identifying the pure tone audiometry to superimpose, or by dragging the desired audiometry from the *session data management bar* in the main area of the graphic interface (see *Maestro* user manual).

Once the test has been transferred to the PC, it can be displayed on a screen other than the live screen. Not only does this screen show the display modes indicated in the *live* window, it can also show:

- The audiogram classification (Merluzzi or MPB)[1] or
- The average of the pure tones (PTA); here, the operator can set the frequencies on which the average is calculated (default frequencies 500 Hz, 1000 Hz and 2000 Hz).

In the case of pediatric audiometry, the screen shows the usual controls used for pure tone audiometry, plus the visual reinforcements panel. For more details, see under the heading *Video-VRA pediatric audiometry*.

The following illustration shows the *live* window for a speech audiometry test.



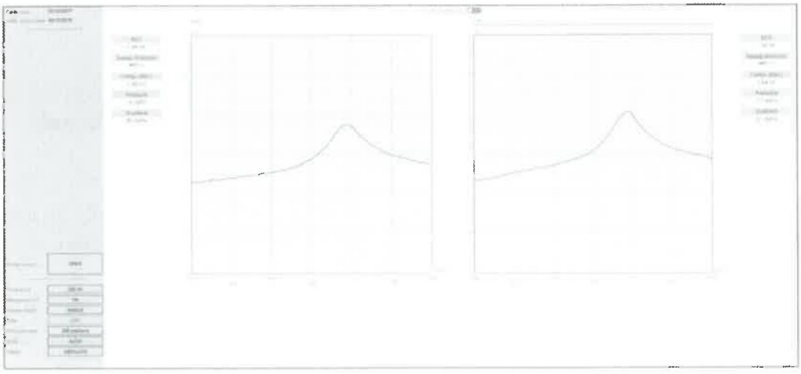
In this instance, the window shows:

- the status of channel 1 (level, side, input, output and frequency);
- the status of channel 2 (level, side, input, output);
- the status of the response button (orange Led lights up when the patient presses the button);
- the test mode (WRS, SRT, MCL, UCL)
- if the selected input is the internal flash memory, the selected material, the name of the current word list and the last word played back;
- the articulation function, which also shows the current percentage of correct answers;
- the speech audiogram / table of thresholds.

---

[1] Enrico Pira, Davide Bosio, Franca Merluzzi: *Prevention of hearing loss due to noise in the workplace – Guidelines proposed by the Italian Society of occupational medicine and industrial hygiene.*

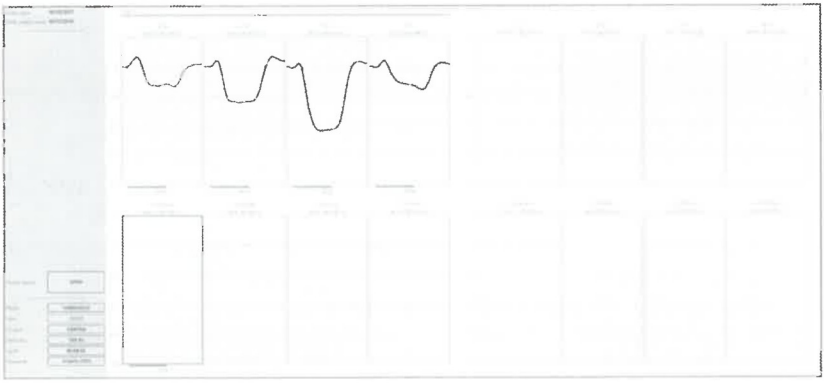
The following illustration shows the *live* window for a tympanometry test.



In this instance the screen shows:

- the probe status;
- the frequency of the probe tone;
- the stimulation side;
- the tympanogram display mode (compensated or absolute);
- the rate at which the pressure sweep occurs during the course of the test;
- the selection of single or multiple scan for the side being tested (*);
- the representation scale;
- the current measurement of the component displayed Y, B, G, B/G;
- the tympanogram and test results.

The following illustration shows the *live* window for an acoustic reflex test.



In this instance the screen shows:

- the probe status;
- the test mode (increasing intensity, fixed intensity, manual, threshold search);
- the transducer (Out), Ipsi or Contra;
- the frequency of the stimulus tone (Stim);
- the intensity of the stimulus tone;
- the representation of acoustic reflexes and test results.

TRANSFERRING TESTS

To transfer tests from an instrument to the Maestro archive, simply select the "TO PC" button on the instrument. Pressing the button when there is a test displayed in the window, only this one test will be transferred; conversely, pressing the button with the main screen of the instrument currently displayed (or rather, the screen showing the list of tests available), all tests conducted on the patient will be sent to the computer.
Once the tests have finished downloading, they can be displayed, and deleted if necessary using the relative archive button.

APPENDIX A:

Troubleshooting

Problem	Possible cause	Solution
Connection between PC and instrument cannot be established	USB port of the computer not working	Connect the instrument to a different USB port
	USB cable damaged	Replace the USB cable (standard USB cable, type A/B)
USB input cannot be selected in speech audiometry mode	Speech material management mode not set for USB	See chapter on instrument settings (audio communication management)
	Speech material not stored correctly on computer	Check that the speech material has been saved to the correct folder. Also, check by way of the ATIT software that the stored speech material is correct
A test cannot be accessed	Optional test not activated	Contact the Inventis technical service department to obtain the license, supplying the serial number of the instrument

"Talk over", "talk back" and "monitor" functions cannot be enabled	Error in audio settings	If "PC" has been selected as the operator-patient communication mode, check that the microphone and headphones are connected to the computer and make certain they are selected correctly in the "Audio settings" window. Alternatively, select "DEVICE" as the operator-patient communication mode.

APPENDIX B:

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